RESEARCH DEPARTMENT

Pulse sound: decoding

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RESEARCH DEPARTMENT

PULSE SOUND : DECODING

Technological Report No. EL-2
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SUMMARY

A system of transmitting television sound by means of position-modulated pulses situated within an extended "back porch" has been proposed.[1] An important aspect of the system is the form of decoder required in domestic receivers and this report recounts work that has been carried out on this aspect of the system.

The report begins with a discussion of the basic processes involved in recovering audio signals from a video signal that includes position-modulated sound pulses. Several experimental decoders are then described which have been developed for addition to a domestic television receiver. The performance achieved by these decoders is described and some indication is given of possible further developments.

1. INTRODUCTION

The proposed system uses a bipolar pulse excursing from black level to white level and to an equal extent in the sync direction. The nominal rest position of the pulse is 7·0 μs after the trailing edge of line sync and peak audio signals deviate the pulse by ±1·5 μs. Research Department Report No. T-181[2] gives a comprehensive specification of the system and Fig. 1 of this report shows a typical line-blanking interval containing a sound pulse.

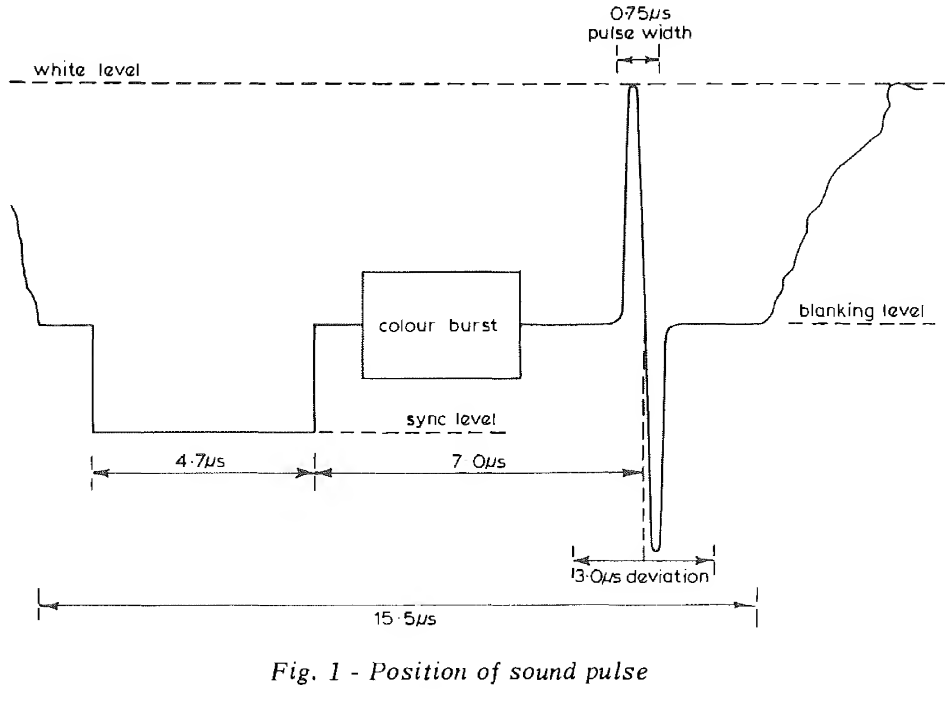

Fig. 1 - Position of sound pulse

The recovery of an audio signal from a video signal containing a position-modulated sound pulse involves three basic processes:

(i) Extraction of the pulse. The sound pulse must be isolated from the remainder of the video waveform and, so far as is possible, from interfering signals superimposed on that waveform; this is necessary in order that operation of the subsequent demodulating circuits shall not be upset by spurious signals. Whenever a spurious pulse reaches the demodulator a sharp crackle occurs in the sound output. Occurrence of such crackles at rates exceeding one or two per second is referred to as "catastrophic failure" of the sound-pulse system.

(ii) Demodulation. The position modulation of the sound pulses must be translated into a voltage or current variation corresponding to the audio signal.

(iii) Filtration. Before transmission, the audio signal is appropriately bandwidth limited, then sampled by pulses at line scanning frequency. In order to recover the original bandwidth-limited audio signal at the receiver, the output from the demodulator must be passed through some form of low-pass filter rejecting all spectral components above half the sampling frequency (i.e. above 7·8 kHz for the 625-line system), as these are entirely spurious.

Each of these processes will now be discussed in more detail.

2. PULSE EXTRACTION

In order to enable the demodulator to function properly, the output of the pulse extraction circuit must fulfil the following conditions:

(*a*) it must consist exclusively of sound pulses;

(*b*) the pulses must form an uninterrupted train;

(*c*) all the pulses must have the same shape;

(*d*) the timing of the pulses must be subject to as little uncertainty as possible.

Techniques for achieving one or more of these objects will now be discussed.

2.1. Time-gating

The deviation range of the sound pulses is $\pm 1{\cdot}5$ μs about their mean position. Any pulse occurring outside this range must therefore be spurious, and in order to afford the maximum protection against interference the decoder must include a gate allowing pulses to reach the demodulator only during the appropriate portion of each line-scan period. However, a time gate which is open for one "n"th of each line period does more than merely reduce the incidence of interference crackles by a factor "n". An additional advantage results from the gate's closure throughout the active-line period, when the effective magnitude of interference may be increased by combination with picture-signal components; thus the gate completely protects the demodulator from this enhanced interference.

In a domestic receiver, the gating waveform is most readily obtained from the line timebase, whose timing typically may vary by one or two microseconds. The resulting imprecision in the point at which the gating period begins can be tolerated, since the video waveform contains no feature other than the colour burst for 7 μs before the mean position of the sound pulse; the colour burst may readily be eliminated by means of a notch filter. The imprecision in the point at which the gating period ends is more serious because the sound pulse, in its latest position, comes within 0·5 μs of the start of the active line. The time gate is therefore liable still to be open at a time when the occurrence of picture signals worsens susceptibility to interfering signals. This difficulty can be overcome by arranging that the extractor, having once delivered a pulse to the demodulator, is thereby rendered incapable of delivering a further pulse for, say, 10 μs. The gating period is thus effectively terminated by the occurrence of the sound pulse.

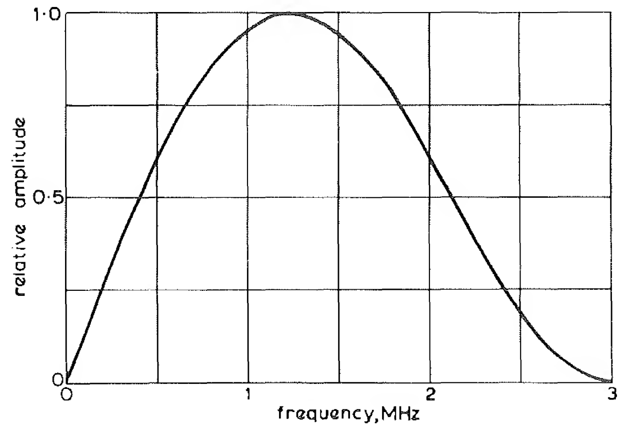

Fig. 2 - Spectrum of sound pulse

2.2. Filtering and Slicing the Pulse

The spectrum of the sound pulse is shown in Fig. 2. It will be seen that the pulse's energy is concentrated within a comparatively small fraction of the video spectrum, centred on 1·35 MHz, and it is evident that a useful degree of discrimination against broad-band interference can be obtained by passing the video signal containing the sound pulse through some form of band-pass filter. The effectiveness of discrimination is increased if the peak

value of the interference is limited to the peak value of the sound pulse before filtering takes place. The combination of peak clipping and filtering affords considerable protection against impulsive interference, and is discussed in a companion report.[3]

In order to obtain the most effective discrimination against interference, it is necessary to exploit the shape of the sound pulse (i.e. its complex spectrum, not merely its energy spectrum). An effective way of doing this is by means of the circuit arrangement illustrated in Fig. 3(*a*). The length of the delay cable, which is driven from its characteristic impedance, Z_0, is such that the time taken for a signal to travel from the driven end to the short-circuited end and back again is equal to the time separation of the peaks in the original pulse. As shown in Fig. 3(*b*) this causes the first lobe of the pulse, reversed in polarity by reflexion at the short circuit, to reinforce the second lobe at the input end of the cable. The circuit is in fact a simple transversal filter, and Fig. 3(*c*) shows its response/frequency characteristic, together with the spectrum of the original pulse. This form of filter is employed in the experimental "professional" decoder that has been used for checking and demonstrating the performance of coding equipment.[2] It

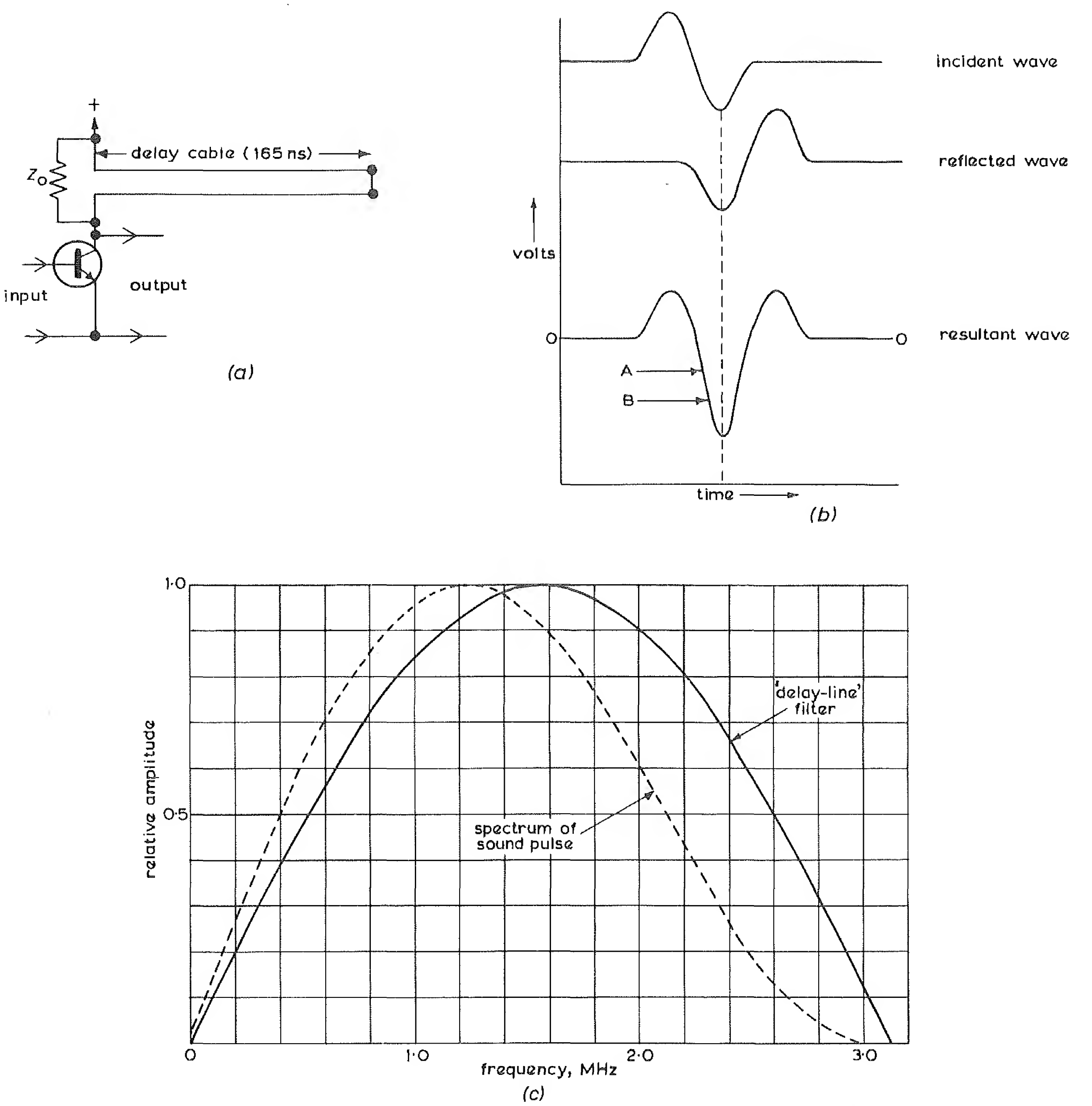

Fig. 3 - "Delay line" filter
(*a*) Circuit (*b*) Waveforms
(*c*) Response/frequency characteristic

has not so far been used in a "domestic" decoder, but its performance serves as a standard for the assessment of other filters.

The position of the filtered pulse may be identified by the moment when the pulse voltage crosses the threshold level of a level-detecting, or "slicing" circuit. This may either by a true "trigger", such as a multivibrator, or a circuit characterized by a very sharp transition from "off" to "on", such as a grounded-emitter transistor having a large collector load.

In discriminating between the spectra of sound pulse and interference, the filter, whatever its type, must not so distort the waveform of the pulse that its position cannot readily be identified. In particular, the filtered waveform should not contain a preliminary lobe that approaches the slicing threshold before the lobe that is meant to cross the threshold occurs; addition of comparatively little interference could then cause this preliminary lobe to operate the level detector erroneously.

Taking the sound pulse shape after extraction as that of the "resultant" waveform of Fig. 3(*b*), the slicing level used for identification of the filtered pulse is chosen as a compromise between conflicting requirements. If signal-to-noise ratio under comparatively favourable conditions were the only consideration, then the slicing level should be set to the level at which the filtered pulse waveform has its maximum slope, say level 'A' in Fig. 3(*b*). This level would, however, be close to zero (that is, to the level obtained before and after the pulse) and spurious operation by interfering signals would occur under less favourable conditions. As the slicing level is progressively moved away from zero, the ability to withstand higher interference levels without catastrophic failure is improved, but the signal-to-noise ratio under favourable conditions becomes slightly worse since the pulse voltage crosses the slicing level with lower slope. Eventually, the point is reached at which large peaks of random noise can prevent a sound pulse from reaching the slicing level, with consequent loss of the pulse. Tolerance to missing pulses is, in general, higher than tolerance to spurious pulses and the optimal slicing level under poor signal-to-noise conditions is therefore that at which, as the video signal-to-noise ratio deteriorates, pulses begin to be lost just before catastrophic failure occurs. This is indicated as level 'B' in Fig. 3(*b*).

A decoder in which slicing occurred at a fixed absolute level would only be suitable for use with a video signal of constant magnitude. In a domestic receiver, however, the magnitude of the video signal is subject to considerable variations, due to aircraft flutter, operation of the contrast control, or the use of a mean-level a.g.c. circuit. Provision must therefore be made for slicing at the preferred point on the filtered waveform, regardless of its absolute magnitude (i.e. the slicing level must be proportional to the pulse magnitude). A form of level detector fulfilling this requirement is the conventional "sync separator" in which d.c. restoration takes place at the base of a grounded-emitter transistor, causing "bottoming" of the collector; this type of circuit is used in the practical decoders to be described later. Although it is, in principle, possible to make a single stage of this type serve as both sound pulse separator and sync separator, the two functions impose mutually incompatible requirements on the design and a satisfactory dual-purpose separator might well be so complex as to wipe out any potential economic advantage.

The output signal from a d.c.-restoring level-detector is not suitable for use as the sampling pulse for demodulation. Primarily this is because the demodulator produces an output that depends upon the epochs of both the leading and the trailing edges of the pulse, and it was found that this type of level detector produces a pulse whose trailing edge is more indeterminate than its leading edge; this observed phenomenon has not been investigated but may be due to the variations in transistor minority-carrier storage resulting from noise-perturbation of the input signal. A further deficiency of the separated sound pulse is that its width is modulated by noise. This precludes its application to a type of demodulator, described later, which derives a reference waveform from the applied pulse by means of linear circuits; the unwanted width-modulation of the pulse then produces unwanted amplitude-modulation of the reference waveform.

It is therefore necessary to interpose between level-detector and demodulator a circuit that initiates, from the leading edge of the detected pulse, a further pulse of unvarying duration and shape. A monostable circuit performs this function admirably; moreover, it can be arranged that the monostable, once triggered, cannot be re-triggered for, say, 10 μs, so that its operation effectively terminates the gating period, as mentioned in Section 2.1.

3. DEMODULATION

Two basically different methods of recovering the audio signal from the separated sound pulses may be used; they are referred to as "Integrating" and "Ramp" demodulation respectively.

3.1. "Integrating" Demodulation

The operation of this form of demodulation is illustrated in Fig. 4. The audio modulating waveform shown in (*a*) is assumed to give rise to the position-modulated pulse train shown by the solid lines in (*b*); the "rest" positions of the displaced pulses are shown dotted. Integration of this wave-

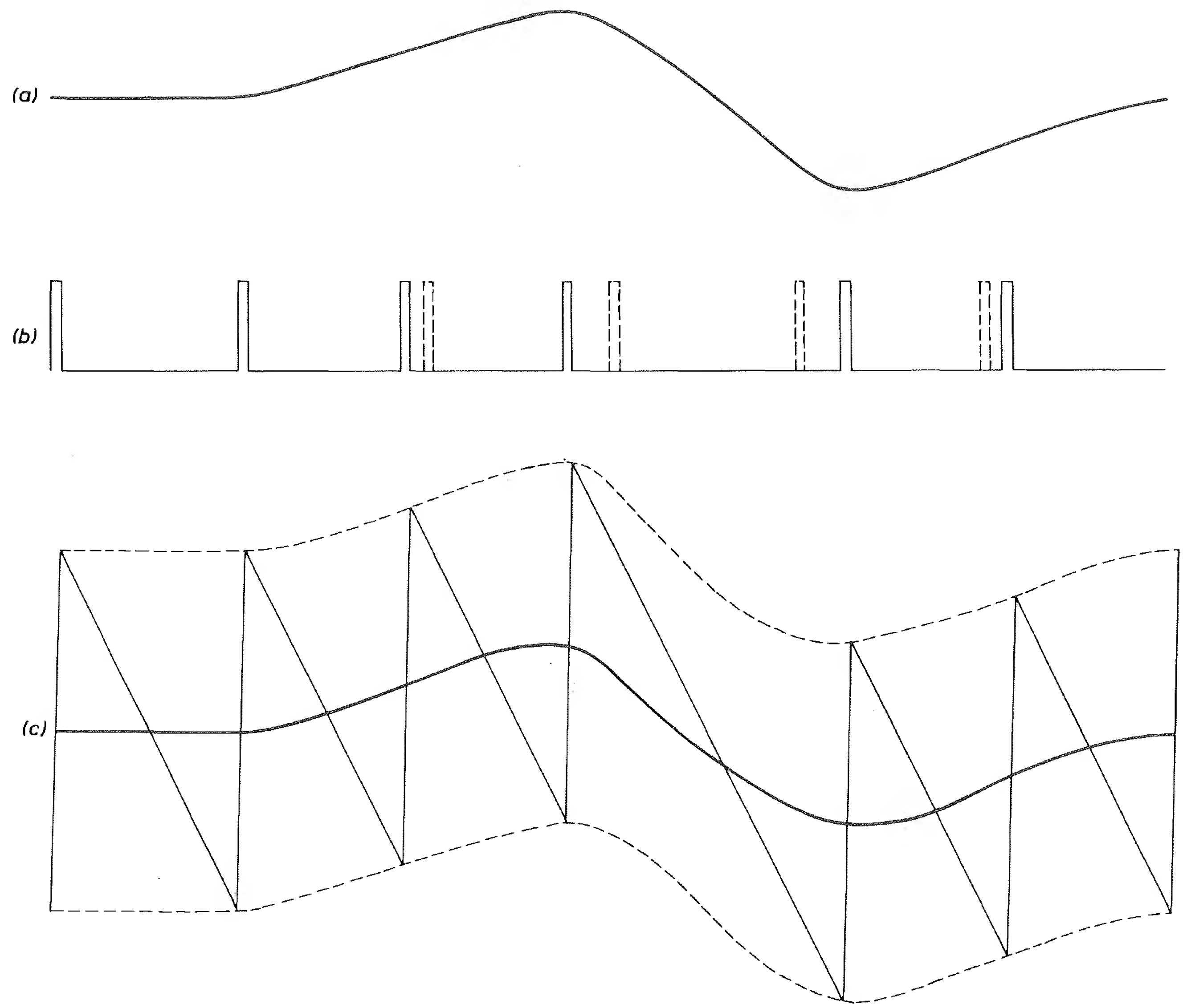

Fig. 4 - "Integrating" demodulation

(*a*) Original audio waveform (*b*) Position-modulated pulses (*c*) Waveform derived by integrating pulses

form (after removal of its d.c. component) yields the sawtooth waveform shown in (*c*). It will be seen that each change of pulse position causes a proportionate change in the envelope and mean value of the sawtooth, which consequently follow the audio modulation. Subsequent removal of high-frequency components by means of a low-pass filter completes the demodulation process.

It is evident that, even with the greatly exaggerated deviation shown in the diagram, the audio signal is of much smaller magnitude than the unwanted sawtooth component, and in practical decoders embodying this principle it is desirable to employ techniques which increase the ratio of the audio signal to the unwanted components. One such technique is described in Section 7.1 of this report. It is also evident from Fig. 4 that if, as a result of noise or interference, isolated pulses were missing from the train of pulses reaching the integrator, each gap in the train would cause a large step in the output of the decoder and hence a severe 'plop' from the loudspeaker. This fundamental property of the integrating demodulator is its main weakness.

3.2. "Ramp" Demodulation

In this method the deviation of the sound pulses is measured by means of an unmodulated reference wave that varies linearly with time during the deviation period of the sound pulses. In Fig. 5, waveforms (*a*) and (*b*) again show an audio modulating waveform and the consequent train of sound pulses. Fig. 5(*c*) shows the relevant portions of the reference waveform, and (*d*) shows the output of a sampling circuit which is actuated by the sound pulse and samples the reference ramp wave.

In a practical decoder, realization of the maximum signal-to-noise ratio requires that disturbance of the reference ramp by random noise shall be negligible; this precludes generating the ramp directly from separated sync pulses. In a domestic receiver the reference waveform can, in principle, be derived from the line-flywheel circuit. However, satisfactory demodulation of sound pulses demands a flywheel of higher performance than is needed for acceptable line-scanning, particularly with regard to disturbances at field frequency. An alternative approach is to derive the reference ramp from the sound pulses themselves. This involves rejection of those components of the spectrum of the pulses affected by modulation; failure to achieve complete rejection of the components corresponding to low modulating frequencies only results in loss of audio response at these frequencies. The required degree of rejection can readily be realized either by a conventional flywheel circuit, or by means of a tuned circuit, and practical decoders employing each of these methods are described later in this report.

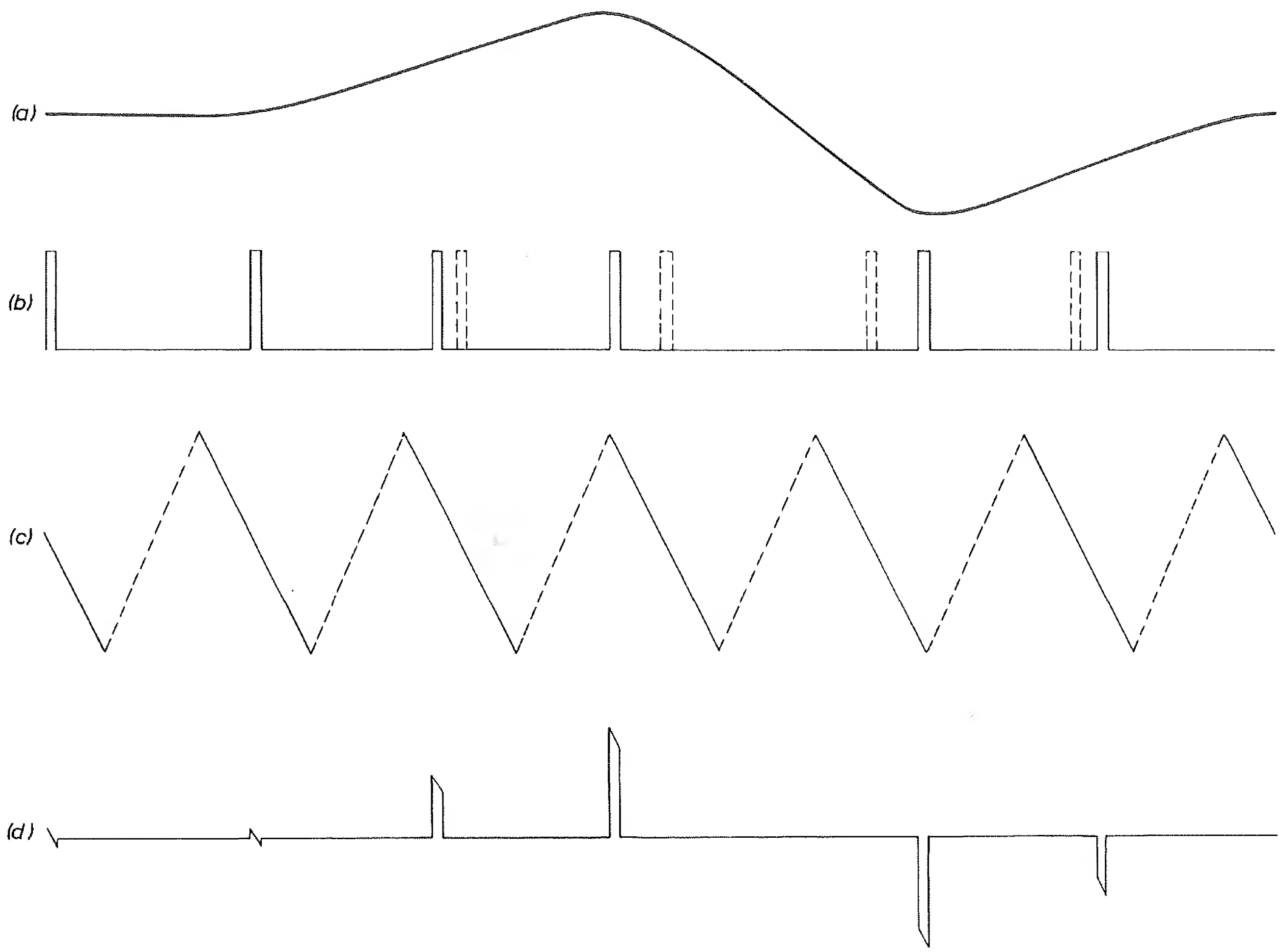

Fig. 5 - "Ramp" demodulation

(*a*) Original audio waveform (*b*) Position-modulated pulses (*c*) Reference ramp (*d*) Result of sampling (*c*) with (*b*)

4. FILTRATION

When the sound pulses are integrated, or used to sample a reference ramp, the original bandwidth-limited audio signal is re-created in the band 0 to 7·8 kHz. However, many other components are present; in particular each audio component of frequency f_a gives rise to an equally strong "image" component at a frequency $(f_s - f_a)$ where f_s is the sampling frequency. As f_a approaches the cut-off frequency, $f_s/2$, the frequencies of the wanted and "image" components approach each other. Therefore, in order to take full advantage of the available audio bandwidth without admitting image components, it is necessary to use a filter having a sharp cut-off at $f_s/2$.

Such a filter is, in fact, used in the "professional" decoder to be described later. For domestic decoders a cheaper filter having a less sharp cut-off must be used, and it is necessary to compromise between maintaining the transmission of wanted high-frequency components within the band 0 to 7·8 kHz and rejecting image components above 7·8 kHz.

No controlled subjective tests have yet been carried out to determine the nature of the best compromise. However, it may be stated here that satisfactory results are obtained with the simple filters used in the "domestic" decoders described later in this report.

In the ramp type of demodulator a measure of filtration can be obtained in the demodulation process itself by using a "sample-and-hold" circuit instead of a straightforward sampling circuit. The filtering effect of such a circuit is to multiply the spectrum of the demodulated signal by a "sin x/x" function having a zero at the sampling frequency, and producing a loss of 3·9 dB at the audio cut-off frequency (i.e. half the sampling frequency). For ramp-type demodulators, the circuit shown in Fig. 6 combines the function of sampling and filtering. This circuit is, in fact, based upon one used in a line-store standards converter[4] where, as here, it is required to smooth a train of pulses at line-scanning frequency into a continuous waveform; it will henceforward be referred to as the "Rainger-Hunt" filter, after its inventors.

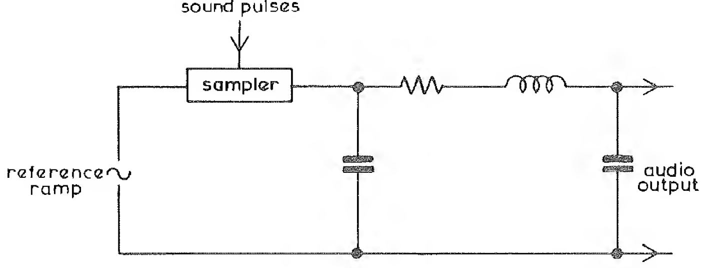

Fig. 6 - "Rainger-Hunt" audio filter

5. MUTING

Under abnormal operating conditions the pulse extractor of a sound-pulse decoder may produce some output other than an uninterrupted train of sound pulses. This is liable to cause all forms of demodulator to produce spurious outputs, usually of much greater magnitude than the proper audio signals, and hence cause distressingly loud noises to issue from the loudspeaker. It is therefore necessary for the decoder to detect the existence of abnormal conditions, and to inhibit the audio output altogether while they persist. This facility is known as "muting".

In a domestic receiver muting may be necessary in any of the following conditions:

Absence of r.f. input:	This can cause the extractor to produce pulses in response to the random noise generated by the receiver.
Gross mistuning:	This can cause the extractor to select irrelevant features of the distorted video waveform in preference to sound pulses.
Loss of line hold:	This produces a large component at the beat frequency between time-base and sync pulses if the line time-base provides either the reference ramp or a time-gating waveform.

Muting, it should be noted, involves inhibiting the audio output throughout the duration of the abnormal condition. Since the waveform feature which causes the circuit to detect abnormality may exist for only a fraction of the time, some form of peak rectification must be used. It is however, necessary to avoid making the peak rectification so effective that an isolated error causes muting. Thus, under adverse but marginally usable conditions of reception, isolated errors occur in the output of the extractor, and in these circumstances isolated noise impulses are preferable to an equal number of periods of silence, each lasting for perhaps a second.

The most appropriate means of providing muting in a given decoder depends very much on the particular circuits used; it is, however, possible to outline the general techniques available.

One method of deriving the required indication of abnormal operation consists of detecting, at the output of the demodulator, any audio-signal magnitude exceeding that produced by peak deviation of the sound pulse.

A second method relies on detecting lack of synchronism between two pulse trains that are normally synchronous; thus, in a ramp decoder in which both the reference ramp and the time-gating waveform were derived from the line flywheel of a receiver, muting actuated by lack of synchronism between the sync-separator output and the line time-base would probably afford adequate protection against abnormal operation.

A third method of muting involves utilizing the mean value of the pulses of current produced by the extractor to generate the d.c. bias required by a subsequent stage. By suitable design of the extractor the number of pulses generated may be made to fall substantially under abnormal conditions, so that this d.c. bias falls and the subsequent stage stops working. A decoder utilizing this arrangement will be referred to as "self-muting", since no semi-conductors are provided specifically for muting.

Simple muting arrangements employing the second and third methods described can sometimes be defeated by particular combinations of picture content, de-tuning, and line-hold adjustment. However, it is doubtful whether a decoder embodying a well-designed extractor and a simple form of muting would fail often enough to justify the additional expense of providing more sophisticated arrangements.

6. SUPPRESSION OF IMPULSIVE INTERFERENCE

In order to simplify the foregoing discussion, impulsive interference has been considered merely as one among several forms of interfering signal. It is, however, a form of interference to which the pulse-sound system is particularly susceptible and, in the absence of specific measures to combat it, impulsive interference spoils the performance of a pulse-sound receiver much more than that of an f.m. receiver, or an a.m. receiver that includes a "rate-of-rise" limiter. A companion report[3] deals with this aspect of the system in detail, and only general principles will be discussed here.

Two distinct methods of rejecting interference impulses have been investigated.

The first method exploits the fact that only one sound pulse is transmitted within each time-gating interval. Therefore, if the pulse-extracting circuit detects more than one pulse in a gating interval at least one of them must be an interference pulse; by suitably inhibiting demodulation whenever this occurs, interference pulses are rejected.

In ramp decoders, demodulation may be inhibited by preventing sampling of the ramp. The effect of doing this depends upon the circuits that follow the sampling point, but in general little noise results during quiet passages, or when the audio signal contains only low-frequency information. Alternatively, the detection of two pulses in one gating interval may be made to restore the output of the sampler to its mean value, which is tantamount to assuming that the missing sample corresponds to a zero in the audio waveform. Again, little noise is added to quiet passages, but in the presence of heavy interference, reproduced speech or music sounds "gritty".

This form of interference suppression is inapplicable to the integrating type of demodulator which, as already mentioned, cannot sustain the loss of a sound pulse without producing a large 'plop'.

Interference suppression by dete ction of more than one pulse in a time-gating interval has been found capable of extremely good results when carefully instrumented in a "professional" decoder. It has not been found possible to apply it with comparable success when instrumentation is subject to the economic limitations of a domestic decoder. One possible reason for this is that in a domestic decoder, the gating period overlaps the beginning of the active line. The suppressor is therefore especially liable to be operated needlessly by interference impulses occurring during this overlap period, since they can be reinforced by combination with the picture signal.

The second method of interference suppression involves reducing the magnitude of the interference pulses to less than that of the sound pulses, at the input of the pulse extractor. The interference pulses will then be totally eliminated if the "slicing level" of the level-detector is above the peak magnitude of interference pulses but still below that of sound pulses. The relative magnitudes of the two types of pulses can be changed in favour of the sound pulse by exploiting the differences in their respective spectra. An interference impulse usually has a bandwidth equal to that of the video signal, whilst that of a sound pulse is appreciably less. If both signals are clipped to the same magnitude, the clipped interference impulse contains less energy within the frequency band of the sound pulse than does the sound pulse itself. Subsequent filtering of the two pulses, therefore, reduces the magnitude of the interference pulse below that of the sound pulse. Suppression of this type is incorporated in two of the decoders described in the next Section.

7. PRACTICAL DECODERS

During the investigation of pulse sound a number of attempts have been made, with varying success, to design "add-on" decoders for domestic receivers,

in order to demonstrate that an acceptable performance is obtainable at a cost comparable to that of the circuits at present associated with f.m. sound. In all of these decoders the sound pulses are derived from the video signal present in the anode circuit of the receiver's video-output stage. This signal is submitted to band-pass filtering, time gating, d.c. restoration and pulse separation, the separated pulses then being applied as trigger pulses to a monostable circuit. This section of the decoder will be referred to as the "pulse extractor".

7.1. Integrating Decoder

As was stated earlier, demodulation by integration produces a very low ratio of wanted signal to unwanted components. In particular, the integrated waveform contains strong components at line-scanning frequency and its harmonics, and it is difficult to reject these adequately with a simple filter; although only the fundamental component is audible, harmonics can cause overloading of the following audio stages.

In the design of a practical integrating decoder, this problem has been eased by lengthening the duration of each sound pulse to half a line period; this can be performed by the monostable circuit provided as part of the pulse extractor already discussed. The pulse train therefore approximates in the absence of modulation, to a square wave, which has no spectral component at twice the line-scanning frequency. This simplifies the filter design since it need only provide (in addition to attenuation between 7 and 15 kHz) a "notch" at the fundamental frequency and adequate attenuation at the third harmonic of line-scanning frequency.

Fig. 7 shows the special integrating circuit used in this design. The perturbed square-wave from the monostable is connected to the bases of the two complementary transistors, and the integrating capacitance, C, is connected to their collectors. By means of d.c. feedback from the collectors of the transistors, the mean collector voltage is maintained at approximately half the supply voltage. The magnitude of the signal from the monostable is such that one or other of the transistors is always cut off. If it is assumed that the monostable produces, from each sound pulse, a negative-going pulse of half-line duration (32 μs) then, following each sound pulse, the p.n.p. transistor charges the integrating capacitor at constant current for 32 μs, after which the n.p.n. transistor discharges it with an equal and opposite current for a period of $(32 \pm \delta)$ μs where δ is the perturbation produced by the modulation.

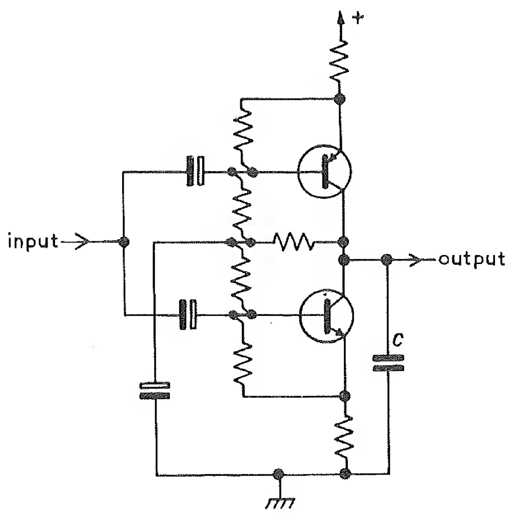

Fig. 7 - "Integrating" demodulator

The filter is of the "RC active" type, and uses one transistor; it is based on a design invented by T.C. Nuttall for a "Cintel" telecine machine, but has been modified to include a notch at line-scanning frequency.

Muting is carried out after filtering, and depends on the detection of audio voltages exceeding those possible in normal operation. The performance of this decoder is, in general, satisfactory. However, as mentioned earlier, an integrating decoder produces a large transient whenever a sound pulse is lost as a result of interference. The slicing level of the pulse separator must therefore be such as to prevent loss of pulses, and this renders the decoder more susceptible to catastrophic failure than one in which similar input circuits are followed by a ramp demodulator. When slicing level in the practical decoder was adjusted for the best compromise, the threshold value of video signal-to-noise ratio,* marking the onset of catastrophic failure, was found to be about 17 dB; in this respect the integrating decoder is 2 dB to 4 dB worse than the other decoders to be described later.

7.2. Decoder Utilizing Receiver Time-Base

An attempt was made to design a decoder in which a ramp demodulator utilized a reference waveform derived from the receiver's own line-flywheel time base. Two principal problems were anticipated. First, it would be necessary to eliminate field disturbance from the time-base waveform in order to avoid hum on the audio signal. Secondly, the filter circuit of the time-base would have to reject low frequencies sufficiently so as to prevent disturbance under noisy reception conditions; the permissible degree of rejection would, however, be limited by the need to preserve an acceptable "pull-in" range.

* Values of video signal-to-noise ratio quoted in this report refer to the ratio of the peak picture signal (black level to white level) to the r.m.s. level of "white" (flat-spectrum) noise.

It is well known that the output waveform of the flywheel line time-base in a domestic receiver is considerably disturbed during each field-blanking interval by the presence of equalizing pulses and broad pulses in the synchronizing waveform. It was therefore arranged to feed the synchronizing waveform to the time-base through a gating circuit which only passed pulses occurring during the receiver's line flyback period.[5] By this means the field disturbance reaching the oscillator through the flywheel filter was reduced to the point where it was no longer the principal source of hum. The hum that remained consisted of a transient during each field-blanking interval, of peak-to-peak magnitude only 25 dB below that of a peak-level audio signal; this was, of course, totally unacceptable. The hum was found to have two components, of comparable severity. One arose from the fact that the suppression of cathode-ray-tube beam-current during field blanking reduced the loading of the E.H.T. supply, thereby altering the working conditions of the line-scan output valve, which in turn affected the frequency of the line-scan oscillator driving it. The second component resulted from electro-magnetic coupling between the field-scanning coils and the transformer of the line-scan oscillator (which was of the blocking oscillator type).

Both these disturbances affected the line-scan oscillator directly, and no improvement was therefore to be gained by elaboration of the flywheel filter. Moreover, they could have been reduced only by elaborating the circuit configuration and changing the layout of the line time-base. This was considered impracticable for an all-valve receiver having virtually all components mounted on a single printed-circuit board. It was therefore decided to abandon this decoder, and attention was transferred to the "sound-flywheel" decoder next to be described. As a result of this decision, it is only possible to speculate upon the difficulty that would have been encountered in designing a flywheel circuit adequate in both its noise rejection and its pull-in range. Experience subsequently gained from the sound-flywheel circuit suggests that a compromise time-base filter could have been designed. However, pull-in performance is impaired by the operation of the gating circuit mentioned above which protects the flywheel from half-line disturbance during the field-synchronizing period. It would probably have been necessary, therefore, to have arranged for this circuit to become effective only after the establishment of synchronism. This could readily have been achieved by utilizing the signal indicating lack of synchronism, which it would in any case have been necessary to provide for muting purposes. However, had it been found impossible to design a time-base filter achieving a satisfactory compromise between noise rejection and pull-in range, it would then have been necessary to use a "bi-modal" filter; in this, a short time-constant prevails until pull-in is achieved, whereupon a long time-constant is substituted. The changeover would again have been effected by means of the signal indicating synchronism. It is however conceivable that a decoder design based upon these ideas might be developed; the picture would benefit to some extent from the more stringent specification of time-base performance imposed by the sound system.

7.3. Sound-Flywheel Decoder

In this form of decoder, illustrated schematically in Fig. 8, a reference ramp is generated by a

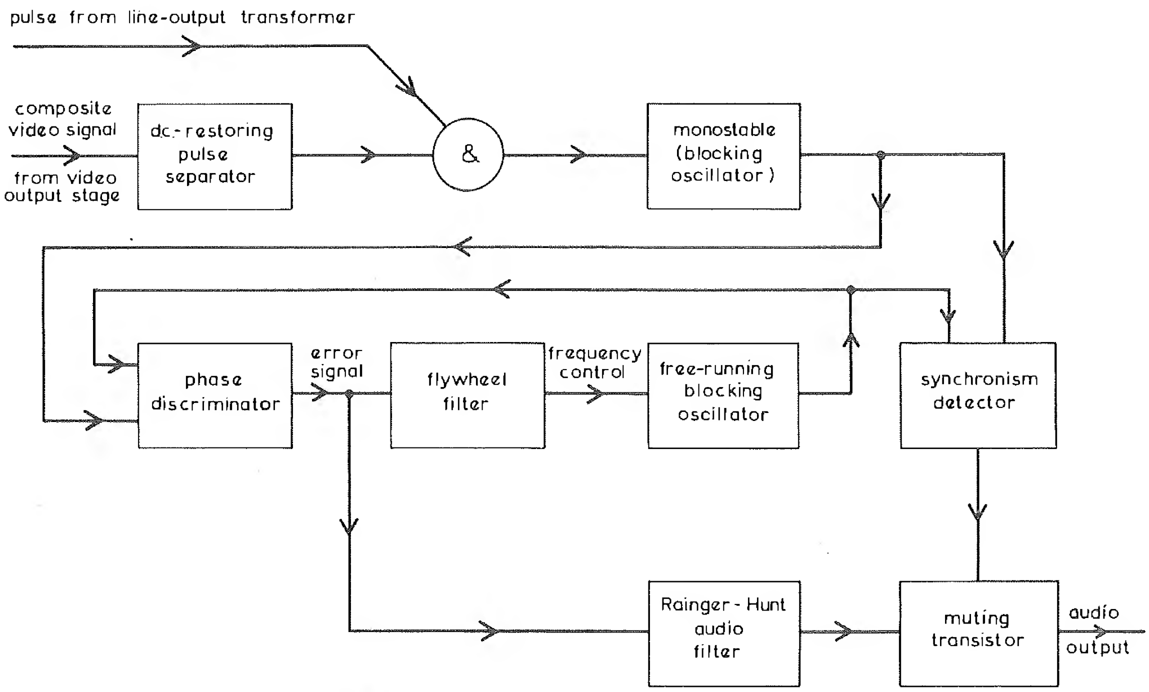

Fig. 8 - "Sound flywheel" decoder - schematic diagram

flywheel circuit that is used exclusively for sound demodulation. This circuit is not supplied with sync pulses, but with separated sound pulses and in its phase discriminator, therefore, each sound pulse samples the steep section of a substantially unperturbed sawtooth. This process is precisely that required for demodulation of the sound pulses, so it is only necessary to pass the output of the phase discriminator through a suitable low-pass audio filter in order to recover the audio signal. Moreover, the sampler can form part of a filter of the Rainger-Hunt configuration, already shown in Fig. 6. A filter of this type was therefore connected to the junction of the phase discriminator and the flywheel filter; the input impedance of the flywheel filter was arranged to be sufficiently high (about 100 kΩ) to avoid significant shunting of the audio filter.

Time gating is achieved by means of a pulse derived from the line-scan output transformer. In the receiver used, additional turns could easily be wound on to the transformer, but in a receiver using a "potted" transformer, it would be necessary to derive the pulse from the line-scan coils.

Muting was obtained by detecting lack of synchronism between the pulses obtained from the monostable and those from the sound-flywheel blocking oscillator. A form of impulsive-interference suppressor (not shown in the diagram) was incorporated in this decoder, in order to demonstrate that comparatively cheap means of suppression could be effective. Whenever the suppressor detected the occurrence of two pulses within the same gating interval, it caused the signal at the input of the audio filter to be momentarily clamped to its mean value. The circuit produced a worthwhile degree of interference suppression but involved the use of two additional transistors. It has since been superseded by simpler and more effective circuits, which depend on preventing the interference pulses from reaching the slicing threshold of the pulse separator.

A disadvantage of this form of decoder is that a "hold" control must be provided for the sound-flywheel oscillator. In the circuit described the pull-in range of this flywheel is about ±150 c/s, and adjustment of the "sound-hold" control would probably be required no more frequently than adjustment of the "line-hold" control.

The performance of this decoder, which contains eight transistors and six diodes, is generally satisfactory; catastrophic failure due to noise occurs at a video signal-to-noise ratio of 15 dB (about 2 dB worse than the figure obtained with a "professional" decoder using the delay-line technique for filtering the sound pulse). Its cost has not been estimated, but is obviously higher than that of the simple 6 MHz amplifier and ratio-detector that are used for demodulation of f.m. intercarrier sound signals in present-day receivers. The circuit, moreover, includes two blocking oscillators, and the cost of each blocking-oscillator transformer must be reckoned as being at least equal to that of an additional transistor.

7.4. "Tuned Circuit" Decoder

This is a ramp decoder which, like the sound-flywheel decoder, derives its reference waveform from the sound pulses rather than from sync pulses; however, it utilizes a tuned circuit in place of a locked oscillator. The tuned-circuit decoder is the most satisfactory of those so far developed; moreover, the practical circuit embodies techniques of pulse separation, time-gating, and muting not covered elsewhere in this report. It is therefore described in some detail in the Appendix. The basic principle involved, however, is described below.

7.4.1. Method of Demodulation

Fig. 9 shows a current generator, which is assumed to generate a train of sound pulses (as shown in waveform 'i'), connected to a high-Q

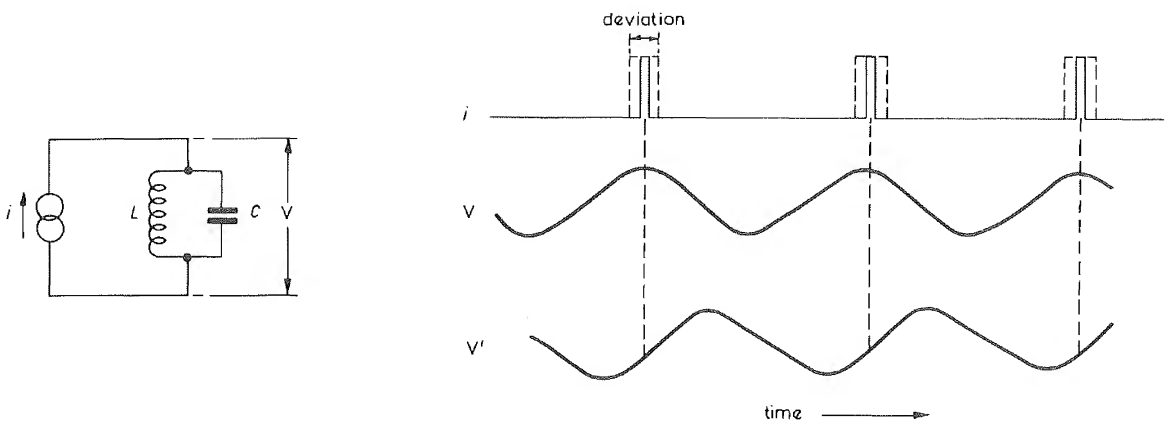

Fig. 9 - Principle of "tuned-circuit" demodulator

resonant circuit. This circuit is assumed to be precisely tuned to the mean repetition frequency of the sound pulses. In the absence of modulation, therefore, the tuned circuit develops a sinusoidal voltage waveform, V, in response to the fundamental component of the current waveform. Applying position modulation to the pulses (represented in Fig. 9 by the dotted lines in the current waveform) generates sidebands, displaced from the mean frequency by multiples of the modulating frequency.

However, so long as the Q of the tuned circuit is sufficiently high to cause effective rejection of these sidebands, the sinusoidal waveform across the circuit will not be disturbed, and can therefore be used for the derivation of a reference ramp. Because the pulse position is deviated by only a small fraction of the repetition period, the sinusoidal waveform itself requires only to be phase shifted by 90° (waveform V′) in order to form a reference ramp of completely acceptable linearity.

The Q of the resonant circuit determines the extent to which the response of the decoder falls as the modulating frequency is reduced, and the sinusoidal waveform begins to follow the phase variations of the input pulse. As will be shown in the next section, however, readily attainable values of Q result in completely adequate low-frequency response.

The practical decoder described in the Appendix uses 6 transistors (one of which would not be needed in a receiver specifically designed for pulse-sound reception), three diodes, and an 18 mm "Vinkor" adjustable inductor.

7.4.2. Performance

The tuned-circuit decoder exhibits better immunity to interference and a better signal-to-noise performance than either of the other practical decoders described. This is primarily due to the performance of the improved pulse-extracting circuits, and not to the particular method of demodulation employed. Catastrophic failure occurs at a video signal-to-noise ratio of 13 dB; this is the same figure as is obtainable with a professional decoder, and is very close to the ratio at which some receivers are rendered unusable by failure of vertical synchronisation. Fig. 10 shows the relationship between video signal-to-noise ratio and audio signal-to-noise ratio* for high and low settings of the contrast control (which gave a 12 dB difference in video signal level). It will be seen that, once video noise becomes the dominant factor in determining audio noise, the audio ratio is some 14 to 15 dB better than the video ratio, depending on the

* Ratio of r.m.s. signal (at a signal level 8 dB below peak modulation) to unweighted r.m.s. noise.

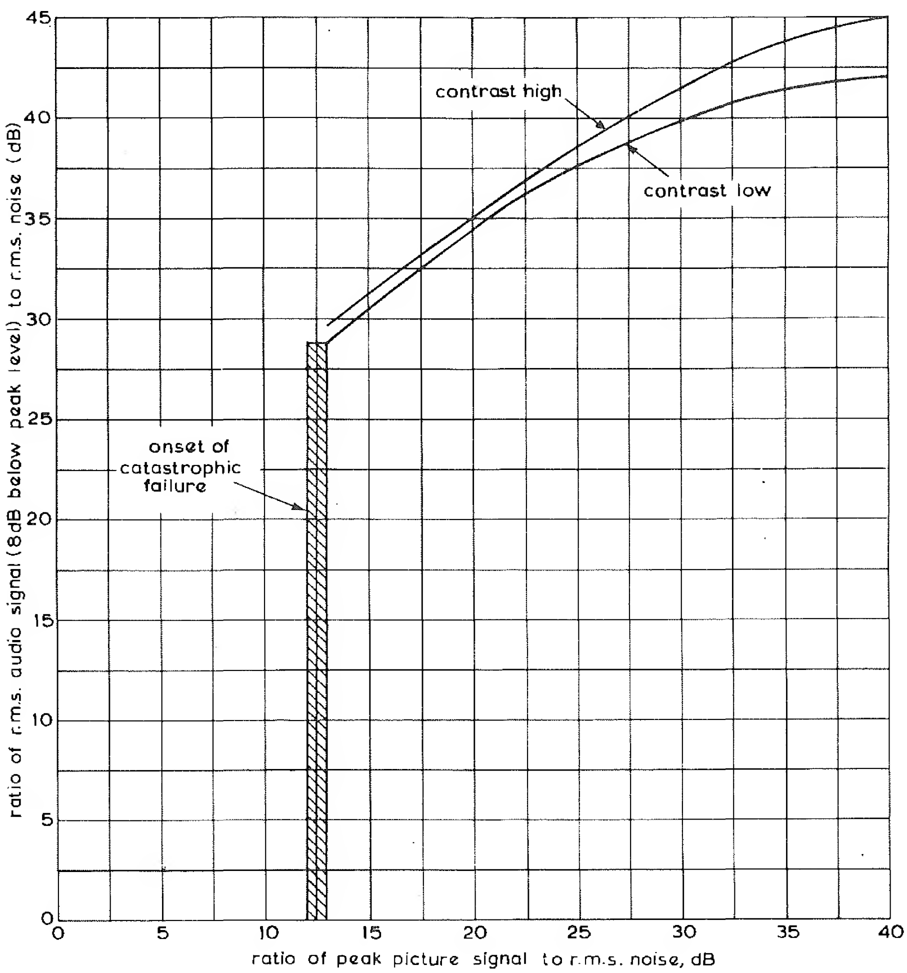

Fig. 10 - Comparison of video and audio signal-to-noise ratios

contrast setting. It is stated in a companion report[2] that a professional decoder, using the delay-line form of sound-pulse filtering described earlier, produces an audio ratio some 15½ dB better than the video ratio. In this aspect of its performance, therefore, the tuned-circuit decoder comes within 1 dB of the professional decoder. Under noise-free reception conditions the audio signal-to-noise ratio reaches a value of 45 dB, which, although poor by professional standards, is quite acceptable for a domestic receiver. The residual noise is thought to be at least partly due to residual modulation in the coder, but has not been thoroughly investigated.

The audio-frequency response is shown in Fig. 11. It will be seen that negligible loss of low audio frequencies results from the finite Q (approximately 130) of the tuned circuit. The high-frequency performance cannot be specified simply by the response attained at frequencies below 7·8 kHz but account must also be taken of the effectiveness with which the spurious components above this frequency are rejected. For a filter of given design, the choice of cut-off frequency determines the compromise that is made between maintaining a uniform response up to the cut-off frequency of the system and rejecting the distortion components beyond it. No subjective tests have yet been carried out on this aspect of decoder design, but the compromise represented by the response shown in Fig. 11 results in sound quality that is completely satisfactory by prevailing domestic standards.

8. GENERAL ASSESSMENT OF THE PRACTICAL DECODERS

The performance of a sound-pulse decoder is very largely dependent upon the circuits preceding the separator stage, which determine immunity to noise and impulsive interference. It has been found possible to design satisfactory circuits using cheap, non-critical components in simple configurations.

In the output end of the decoder, both the "Rainger-Hunt" filter and a simple RC active filter have been found capable of providing adequate filtering at low cost.

It is in the development of demodulating circuits that there is most room for improvement. The integrating demodulator is basically simple but does not withstand adverse reception conditions as successfully as a ramp demodulator. The tuned-circuit demodulator has a good performance and uses only a modest number of components, but includes an inductor which must combine high Q and high stability. Moreover, it is necessary to provide a pre-set control for adjusting either this inductor or its associated capacitor, which would cost more than the provision of a pre-set variable resistor.

The tuned-circuit decoder described in the Appendix has only three points of contact with the receiver - "supply volts in", "video in", and "audio out". It therefore offers the prospect of producing an 'add-on' decoder that would be suitable for virtually any 625-line receiver. This might allow some abbreviation of the transition period preceding a final changeover to the pulse-sound system, since it would no longer be necessary to delay changeover until all present-day receivers had been discarded.

9. FUTURE DEVELOPMENTS AND CONCLUSIONS

The work described in this report represents a comparatively small expenditure of effort by engineers with no experience in designing domestic receivers, and has been limited to the development of 'add-on' decoders for existing receivers. Considerable economies in design would be expected to result if engineers more experienced in designing cheap reliable circuits were set to work on the problem, especially if they were able to develop all circuits following the video detector with pulse-sound transmissions in mind. The circuits so far developed have, however, established the feasibility of simple decoders.

As an alternative to the use of a high-Q tuned circuit, it should be possible to develop a decoder which depends upon the same principle of sideband rejection as the tuned-circuit decoder, but which utilizes instead some form of RC active circuit exhibiting resonant behaviour. Such a development would be likely to increase the number of semiconductors required; however, it is possible that before the earliest date at which a changeover to pulse sound could be completed, the application of integrated-circuit techniques to domestic receivers would have made this a desirable exchange.

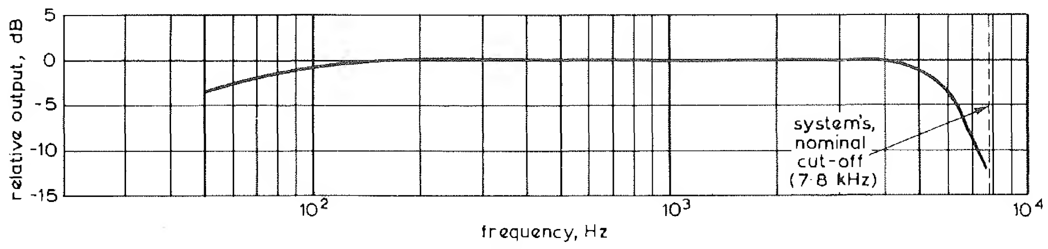

Fig. 11 - Frequency response of tuned-circuit decoder

The use of two independent line-frequency flywheel circuits in one television receiver is, in principle, wasteful. It could therefore be argued that any development of a receiver specifically for sound-pulse reception should include the development of a flywheel capable of serving both as line time-base and as ramp generator. Such a flywheel would have to be driven by sync pulses and include means for eliminating field disturbances; it is not practicable to drive a dual-purpose flywheel with sound-pulses, because an impracticably narrow noise-bandwidth is required if the lowest audio frequencies are not to produce unacceptable disturbance of the raster.

There is a strong case, however, for the use of two separate flywheel circuits, even in receivers designed specifically for sound-pulse reception. A sound flywheel can be considerably disturbed by low audio frequencies without any ill effects beyond a slight fall in l.f. response, whilst conventional line-flywheel circuits can, and do, suffer considerable field disturbance with negligible detriment to the subjective acceptability of the picture.

The designer must therefore provide either one flywheel of immaculate performance or two flywheels which may be of relatively mediocre performance.

10. REFERENCES

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5. Provisional Patent Application No. 12880/66.

APPENDIX

Details of a Practical "Tuned-Circuit" Decoder (Fig. 12)

Interference Suppressor and Pulse Extractor

The video input to the decoder is derived from the cathode of the receiver's cathode-ray tube by means of a matching and inverting transistor (not shown). Level detecting occurs at the base of TR1, after the functions of impulsive-interference suppression and pulse filtering have been performed by the simple circuit shown.[3]

The video signal is clipped by the diode, D1, and then filtered by the series-resonant circuit (100 μH, 100 pF), whose resonant frequency is roughly that at which the sound pulse contains maximum energy. As shown in the sketch waveforms, this arrangement has been found to produce, at the input to the level detector, sound pulses which exceed the magnitude of interference pulses sufficiently to ensure that, under most practical conditions, only the sound pulses cross the slicing level of the level detector.

The pulse-extracting circuit utilises diodes for d.c. restoration, and so does not require a separating transistor to precede the monostable circuit (TR1 and TR2). However, when d.c. restoration is combined with triggering, it is necessary to ensure that the abrupt change of conditions occurring when the monostable fires does not affect the process of d.c. restoration (which ensures that the sound pulse is sliced at the correct level despite changes in the magnitude of the video signal). In the present circuit, this problem is overcome by using two d.c. restoring diodes (D2 and D3). At the beginning of the sound pulse, the "anode" potential of D3 is slightly less positive than that of D2, which is held at the "bottomed" base voltage of TR1. The negative-going pulse thus causes D2 to conduct. As soon as the current drawn is sufficient to trigger the monostable, however, the base potential of TR1 falls by several volts, thus cutting off D2, and the remainder of the sound pulse is d.c. restored by D3.

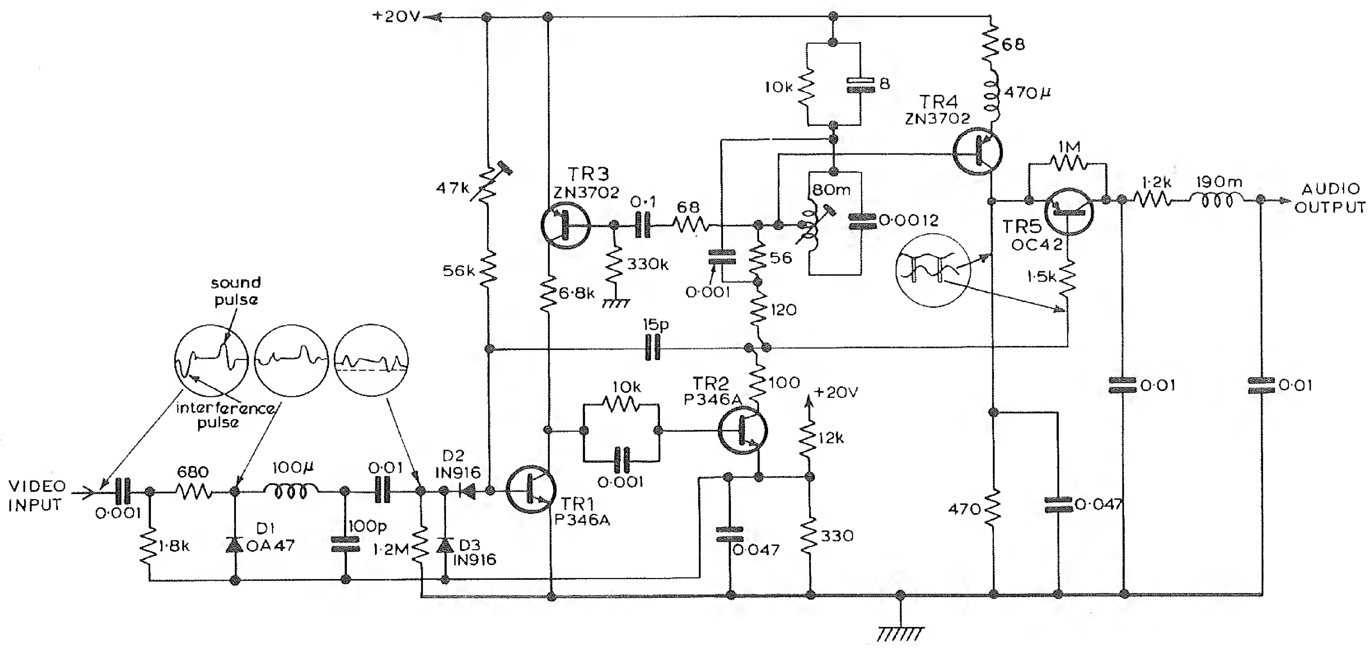

Fig. 12 - Circuit diagram of tuned-circuit decoder

It will be noted that D3 "anode" is not connected to a fixed potential, but to a potential that rises during the unstable period of the monostable, as a result of the current drawn by TR2, and falls during the succeeding line period. In this way, noise pulses occurring at the end of the time-gating period (i.e. during the first microsecond or two of the active line period) are prevented from re-triggering the monostable, since they cause D3 to conduct, not D2. Although this expedient prevents noise pulses from re-triggering the monostable, it increases the probability that they will cause D3 to conduct, to the detriment of the d.c. restoration. This undesired side effect is prevented by arranging that the sawtooth potential is also supplied to the components preceding the d.c.-restoring capacitor.

The monostable circuit generates pulses whose duration (adjustable by means of the variable resistance in the base circuit of TR1) is about $\frac{1}{2}$ μs. If a longer value is used, the monostable pulse can, for deviated positions of the sound pulse, extend into the active line period. The precise moment at which TR1 conducts and the monostable pulse terminates, can then be disturbed by high-frequency components of the picture signal, which break through the capacitance of the diode D2.

Demodulator

In the interests of economy, the L/C ratio of the tuned circuit has been made high enough to allow a low-loss capacitor of modest capacitance ($0{\cdot}0012\ \mu$) to be used. This necessitates the use of a tapped inductor, in order to prevent the Q of the resonant circuit from being substantially reduced by the other circuits connected to it; the number of turns between the tap and the earthy end of the coil is about one twentieth of the total, giving an impedance transformation of about 400 : 1. In order to prevent the leakage inductance of the tapped coil from upsetting the operation of the monostable circuit, it has been found necessary to decouple the tuned circuit from the monostable at high frequencies.

The waveform across the tuned circuit is amplified and phase-retarded by TR4; the required retardation of 90° is achieved partly by means of the inductor in the transistor's emitter circuit, and partly by the use of a parallel RC combination as its collector load.

The d.c. conditions are so arranged that the switching transistor, TR5, is turned on during each monostable pulse but at no other time. This is illustrated in Fig. 12 by the encircled sketch showing the relative d.c. levels of "switching" and "sampled" waveforms.

Time Gating

Time gating is achieved by means of the transistor TR3, which acts as a switch controlling the application of collector supply-potential to TR1. The monostable can thus be fired only when TR3 is conducting. The base circuit of TR3 is connected, by means of a d.c.-restoring coupling, to the tapping on the tuned-circuit inductor, and the transistor is thus turned on only during the negative peaks of the sinusoidal waveform. When the resonant circuit is correctly tuned, the conduction period of TR3 is symmetrically disposed about the rest position of

the sound pulse; this condition, in fact, affords a convenient means of checking the tuning of the resonant circuit. When a video waveform is first applied to the decoder no sinusoidal waveform exists across the resonant circuit, and so no gating can take place. The base resistor of TR3 is therefore returned to earth potential in order to ensure that the gate is held "on" until the sinusoidal waveform builds up.

Muting

The method of muting employed depends on the fact that abnormal operating conditions generally result in a substantial reduction in the mean rate of firing of the monostable circuit. In its stable state the monostable has TR2 cut off, and the mean current drawn by this transistor is thus proportional to its rate of firing. This mean current develops a direct potential across the 8 μF electrolytic capacitor in its collector circuit, which in turn provides the d.c. drive for the following transistor, TR4. This drive is dependent upon the duration of the pulse from the monostable circuit, and a variable resistor in the base circuit of TR1 allows the correct operating condition of TR4 to be pre-set. Under abnormal conditions the drive decreases substantially, and the collector potential of TR4 falls to such an extent that it is at all times more negative than the switching potential applied to the base of TR5, which consequently remains permanently cut off; thus no sampling occurs and no audio signal is developed. The high resistance bridging the switching transistor, TR5, ensures that the same mean potential exists on each side of the transistor; the value of resistance is too high to allow significant break-through under muted conditions, or to interfere with normal operation of the decoder.

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